Lecture 6

Pulse Modulation

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TIP
For additional advice see
Dale Carnegie Training®
Presentation Guidelines

Contents

- Sampling Theory.
- Pulse Amplitude Modulation PAM.
- PPM and PDM.
- Delta and Adaptive Delta.
- PCM.

Digital Importance

- Simplicity of digital circuit design,
- Ease of applying IC Technologies,
- Widespread use of computers in handling all forms of data,
- Using of coding to minimize effects of noise and interference,
- Using of digital signal processing, compression, scrambling, …etc.

Sampling Theory

Sampling Signal

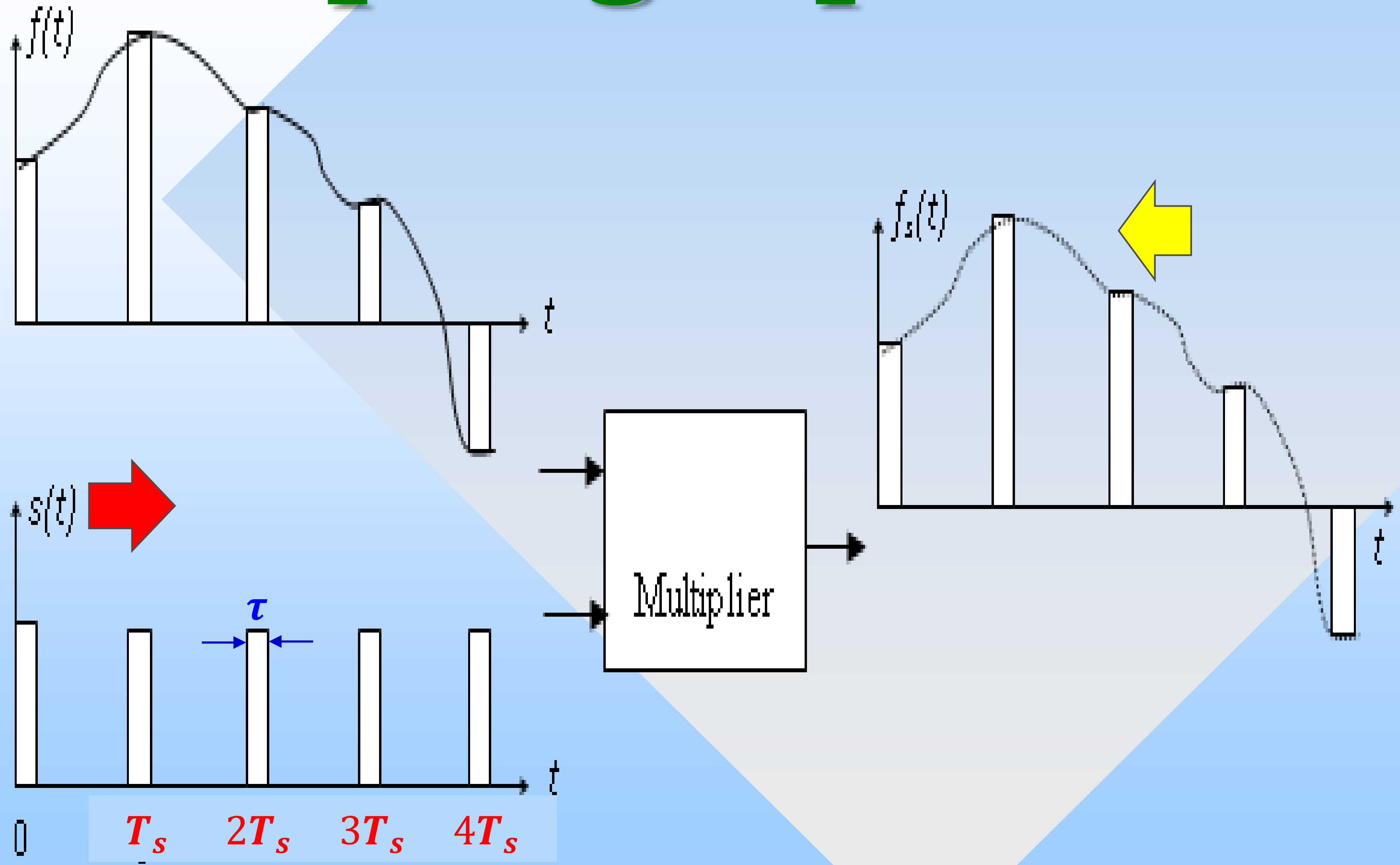
A periodic pulse train expanded by Fourier series to:

$$s(t) = \frac{\tau}{T_s} + \frac{2\tau}{T_s}(\cos \omega_s t + \cos 2\omega_s t + \cdots)$$

$$\omega_s = \frac{2\pi}{T_s}$$

$$s(t) = \frac{\tau}{T_s} + \frac{\tau}{T_s}\left(2\cos \frac{2\pi}{T_s} t + 2\cos 2\frac{2\pi}{T_s} t + \ldots\right)$$

Sampling Operation



Sampler Output

- The sampler (product output) will be:

$$f_s(t) = f(t)s(t)$$

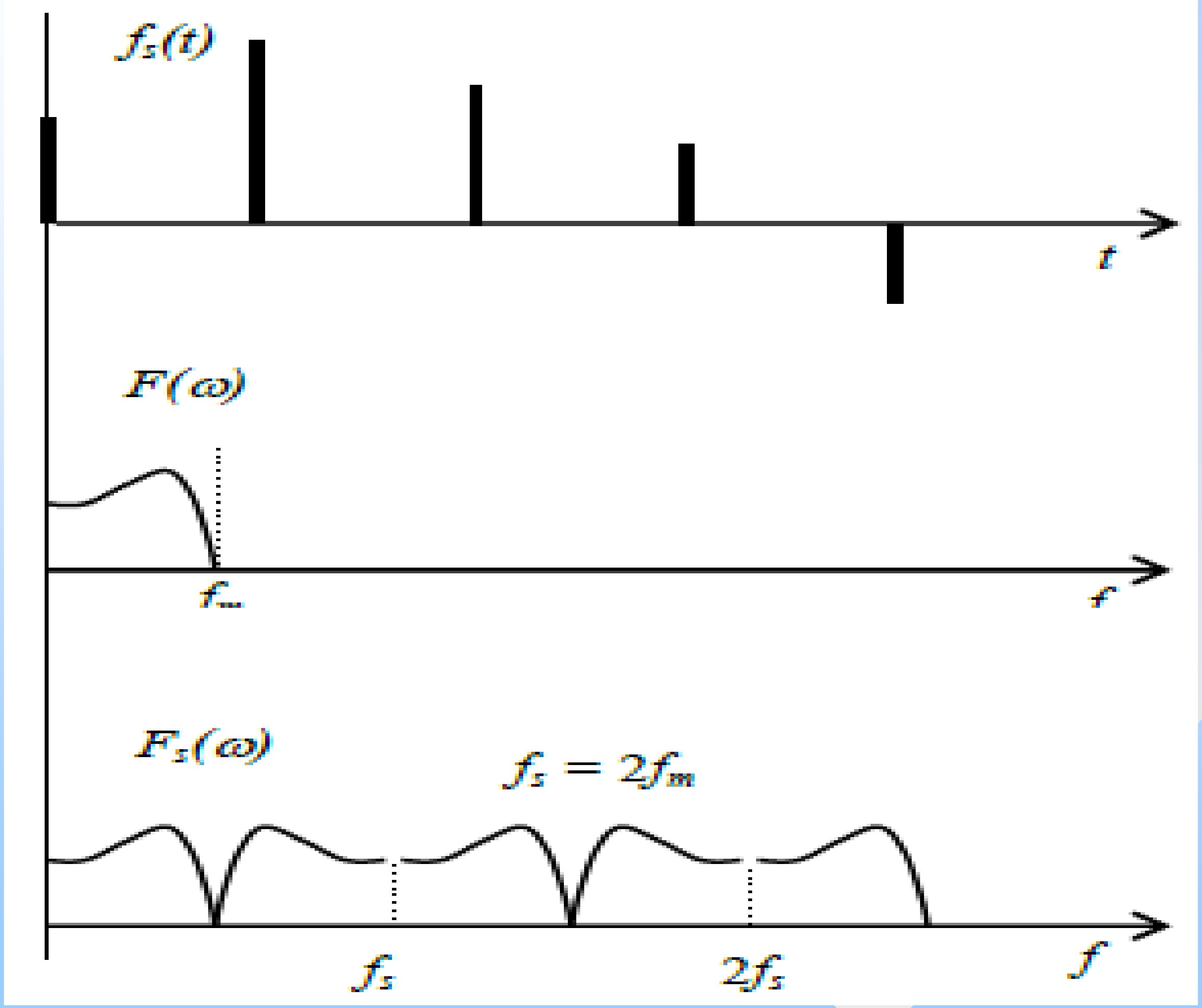
$$f_s(t) = \frac{\tau}{T_s} f(t) + \frac{\tau}{T_s}\left(2 f(t)\cos\frac{2\pi}{T_s}t + 2 f(t)\cos 2\frac{2\pi}{T_s}t + \dots\right)$$

- Assume $f(t)$ baseband signal of maximum frequency f_m and $T_s = 1/2f_m$ the output is:

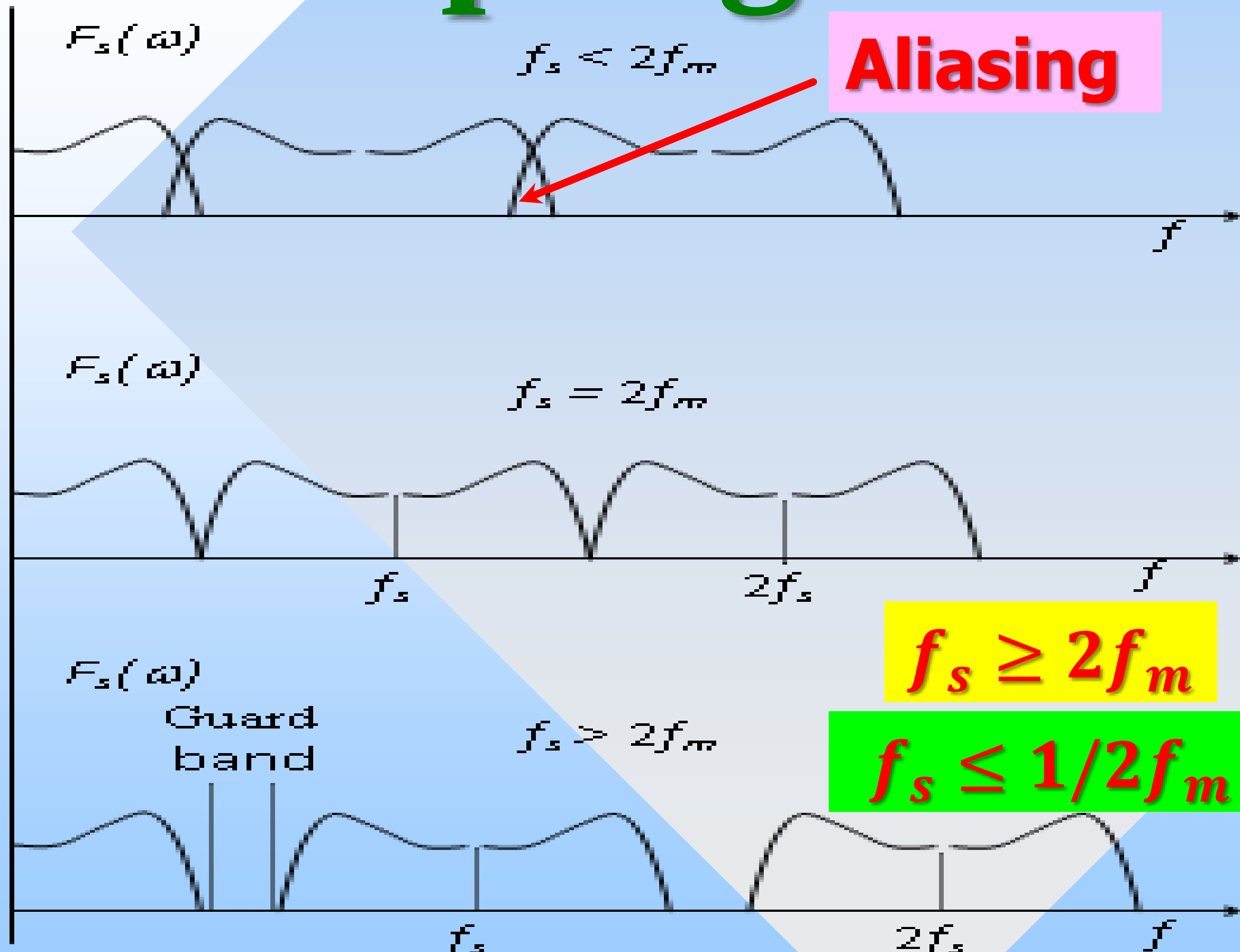
$$f_s(t) = \frac{\tau}{T_s} f(t) + \frac{\tau}{T_s}(2 f(t)\cos(2f_m)t + 2 f(t)\cos(4f_m)t + \dots)$$

$$f(t) \quad + \quad DSB.SC_{\,2f_m} \quad + \quad DSB.SC_{\,4f_m}$$

Sampler Spectrum



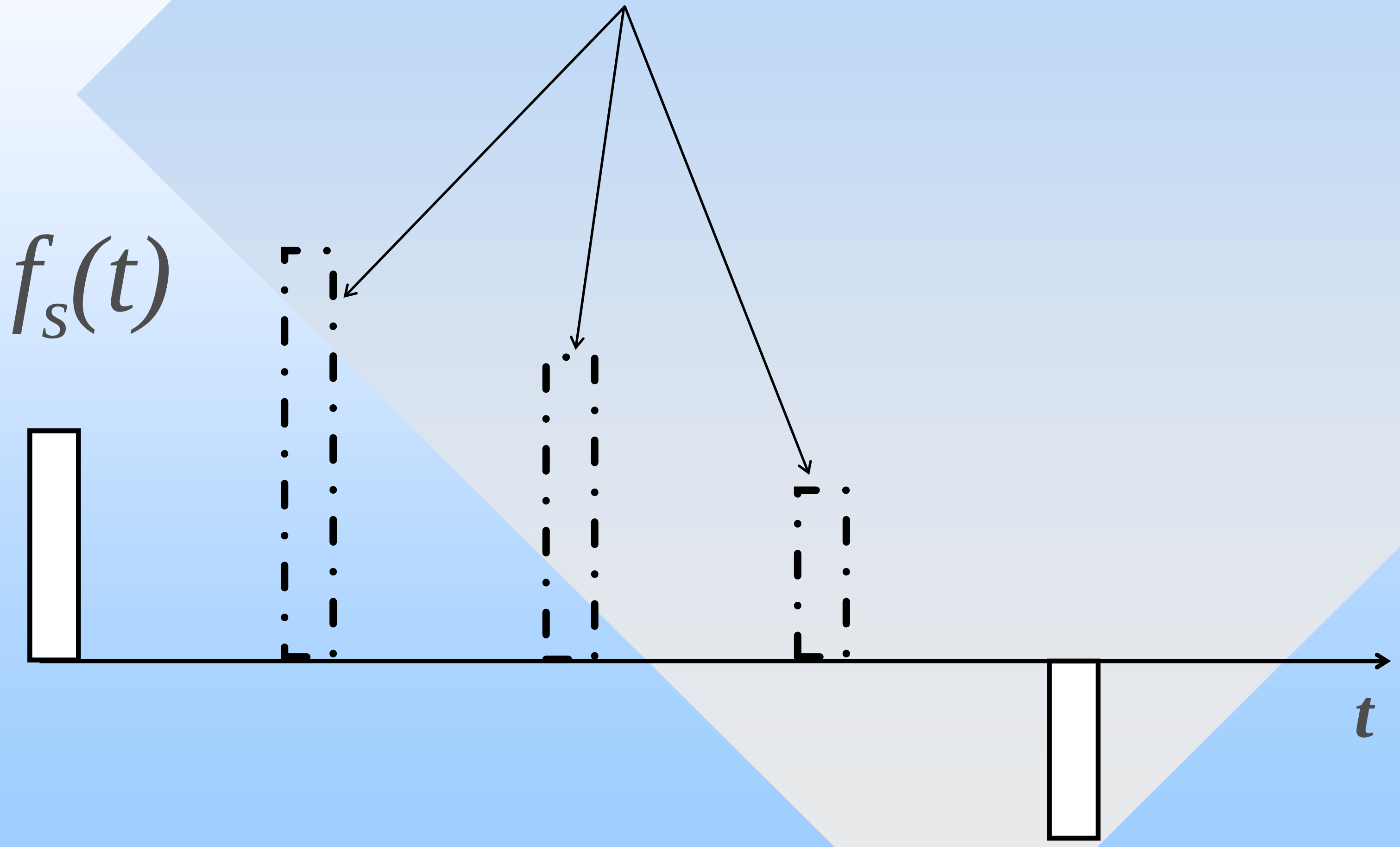
Sampling Rate



$$f_s \geq 2f_m$$

$$f_s \leq 1/2f_m$$

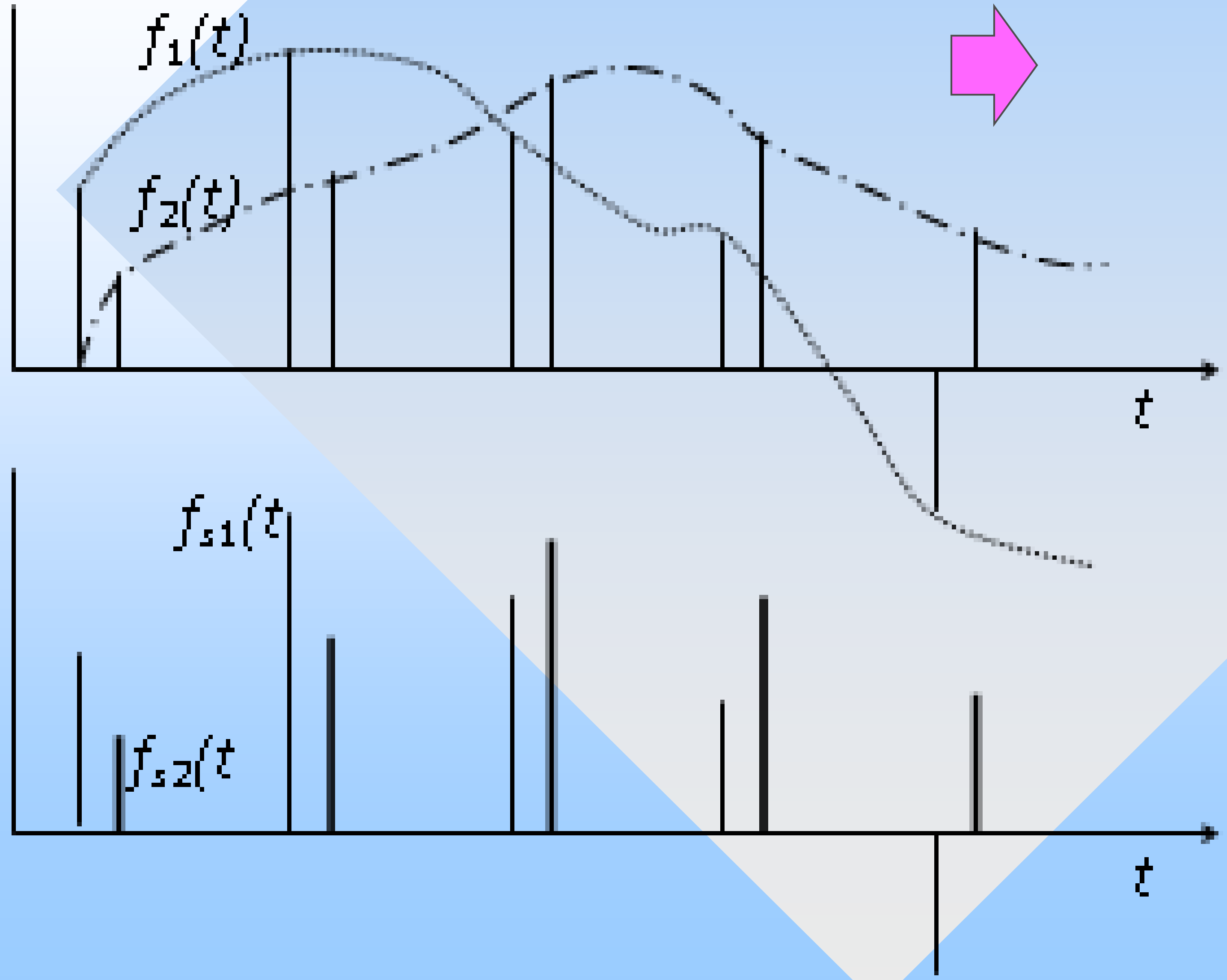
These samples should be added



Types of Sampling

- Instantaneous Sampling:
  - Hardly feasible.
  - Samples at transmitting end have infinitesimal energy
  - Received samples will be infinitesimally small
- Natural Sampling:
  - Samples top take the shape of original signal.
  - Received samples can not maintain its shape.
- Flat Top Sampling:
  - It has the merit of simplifying design.
  - It introduces some distortion.

Instantaneous Sampling



Natural Flat Top

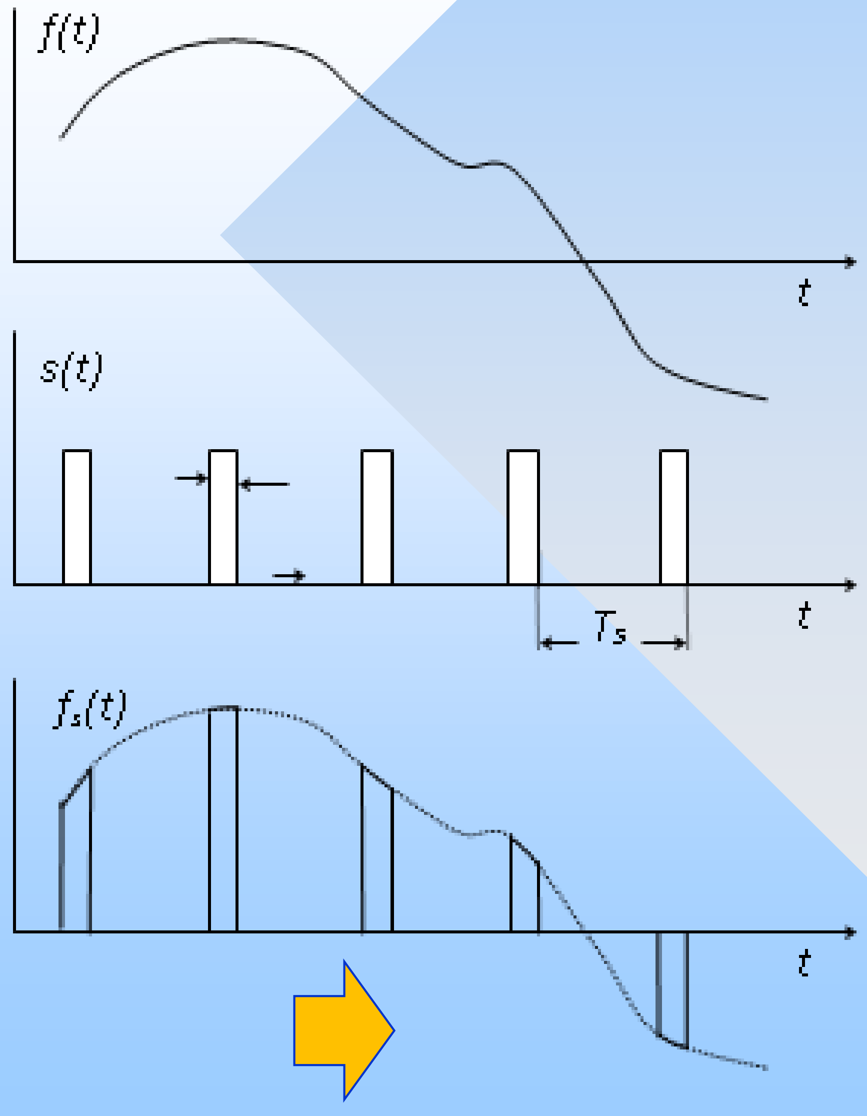


Fig.1.7 PAM (Natural Sampling)

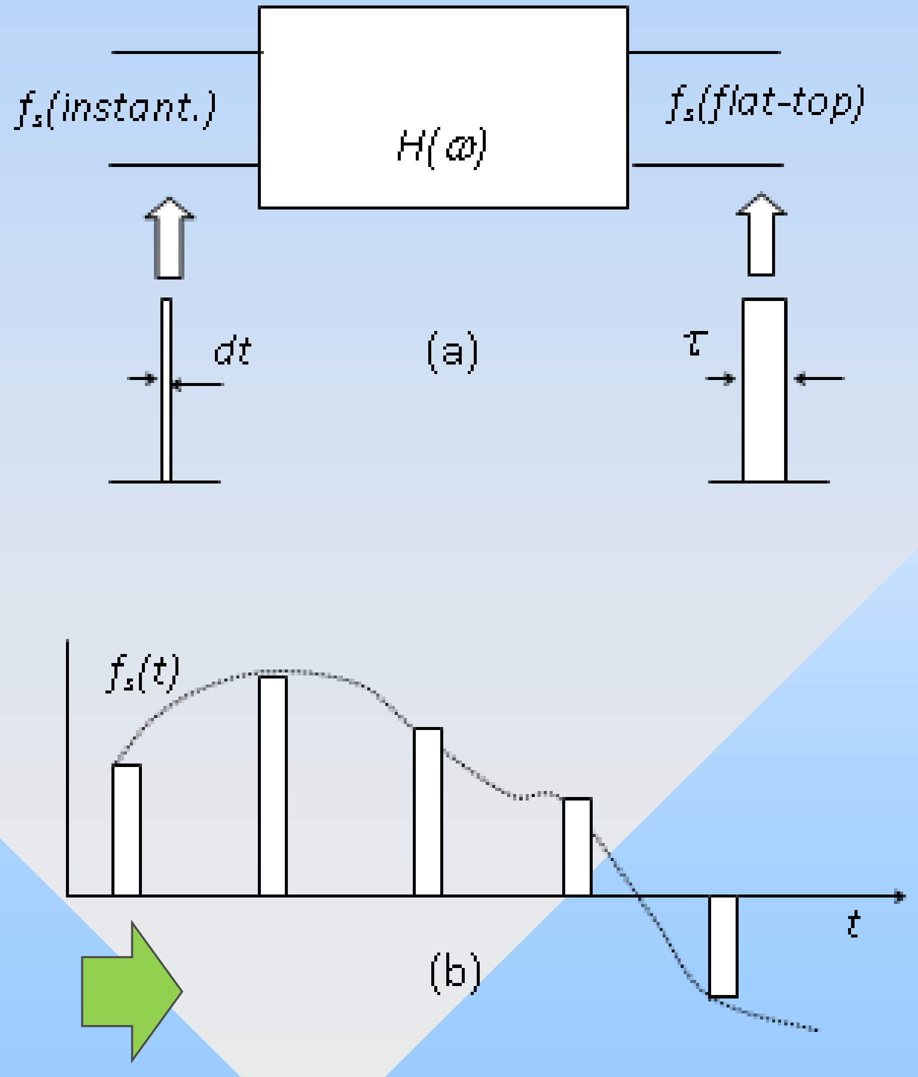
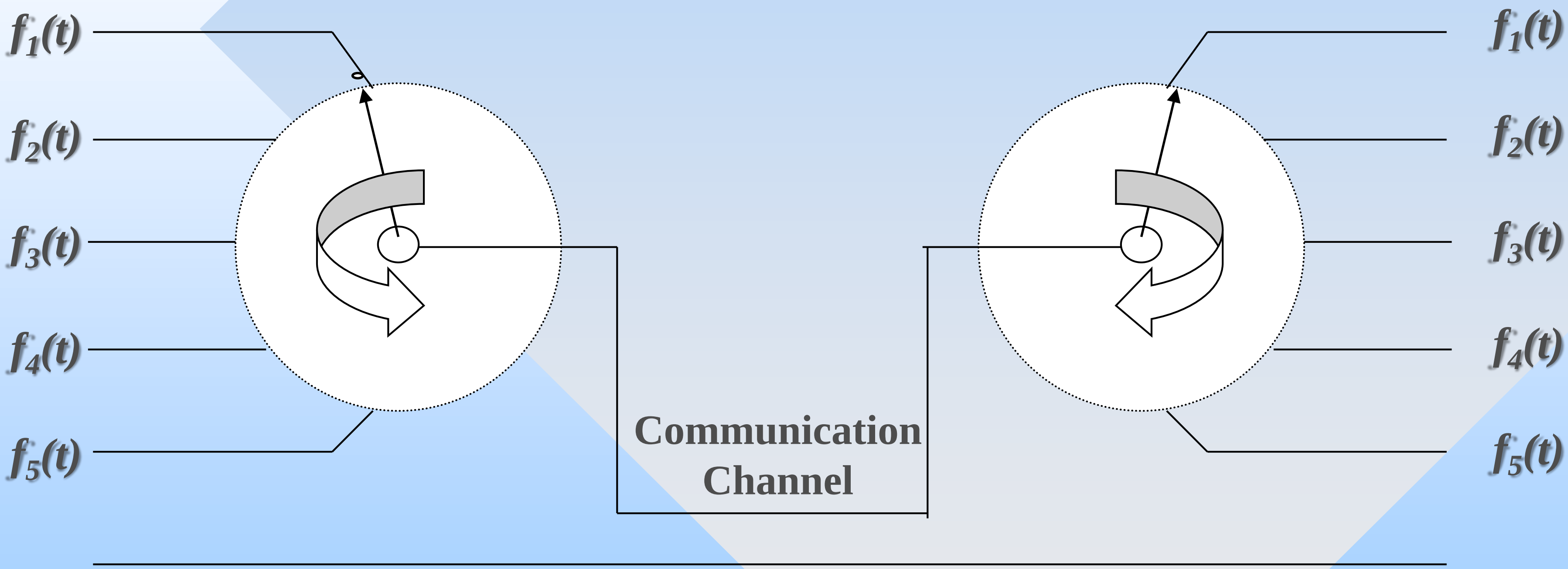


Fig.1.8 PAM (Flat-top Sampling)

PAM

Pulse Amplitude Modulation

Time Division Multiplexing and Sampling Principle

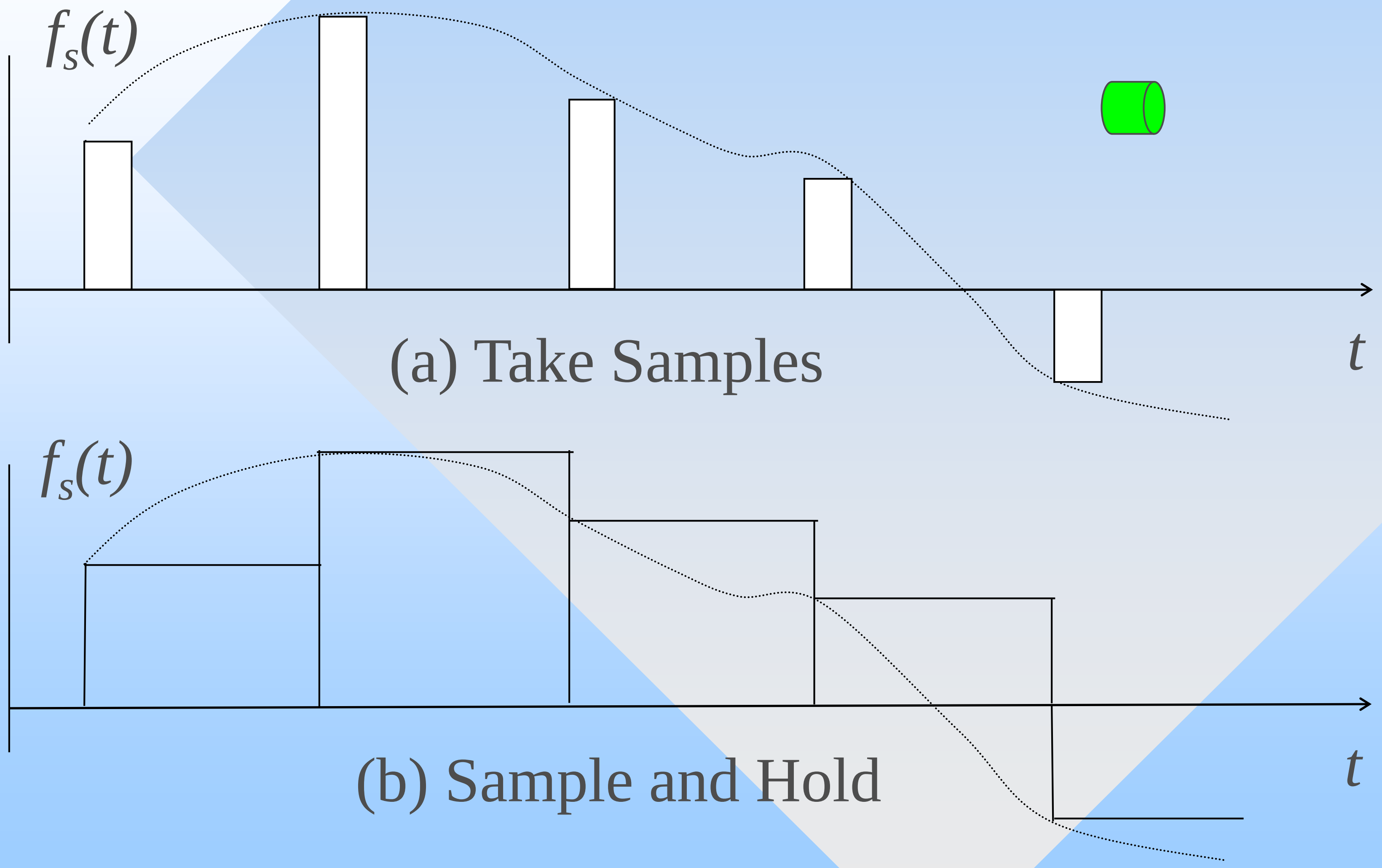


Sample and Hold

Assume that N signals are being time division multiplexed TDM:

- As N increases, $\tau/T_s = 1/N$ becomes smaller and so does the output signal.
  - One method is to use amplifiers which may introduce noise.
  - Or sample pulses are extended or hold until the occurrence of the next sample.
- It has the advantage of filtering being adequate,
- Disadvantage that some distortion must be accepted.

Sample and Hold



Recovery of PAM

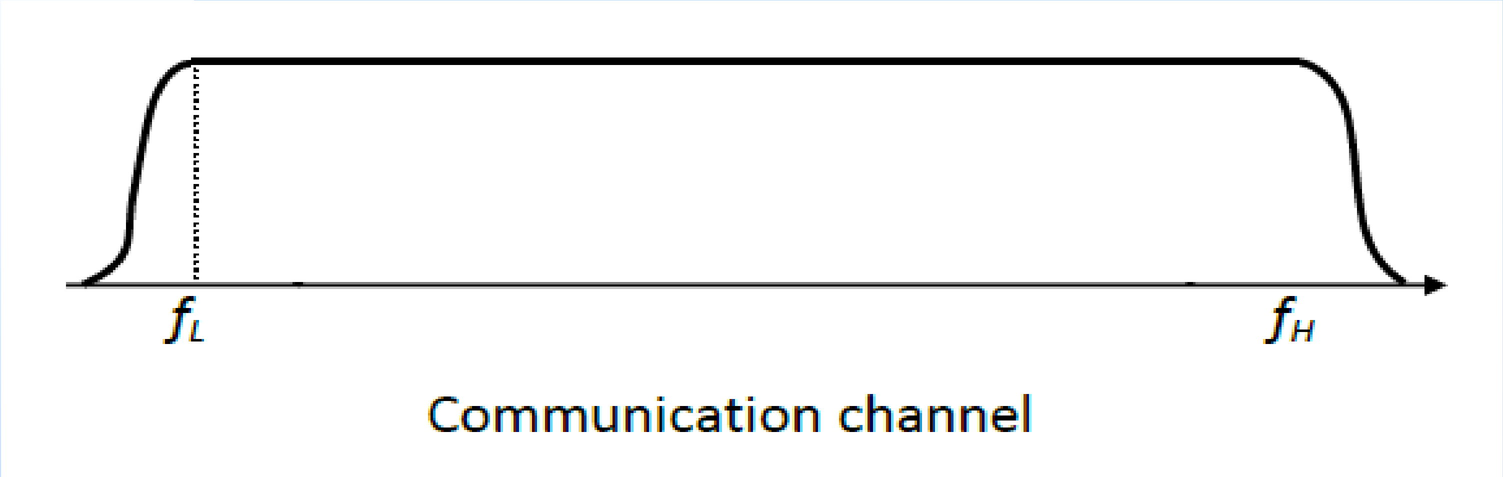
- Using LPF:
- Using sample & hold circuit and LPF

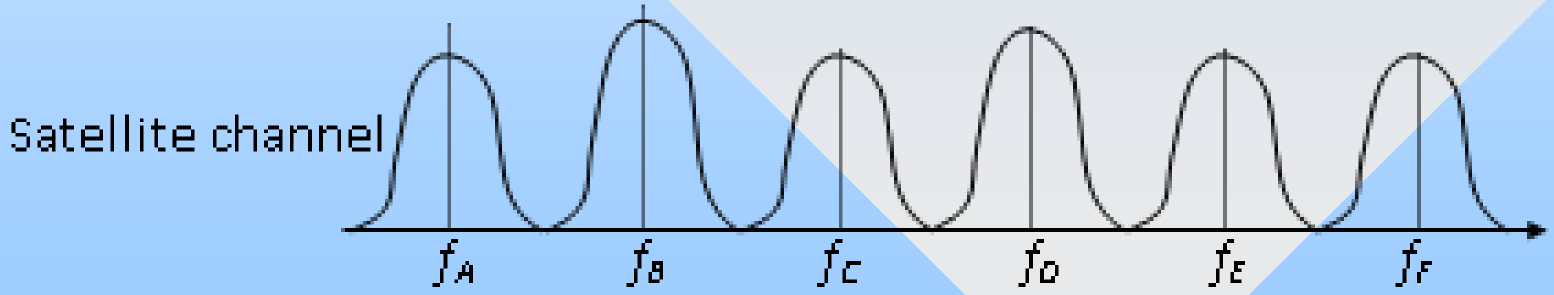
Crosstalk

Reason of Crosstalk

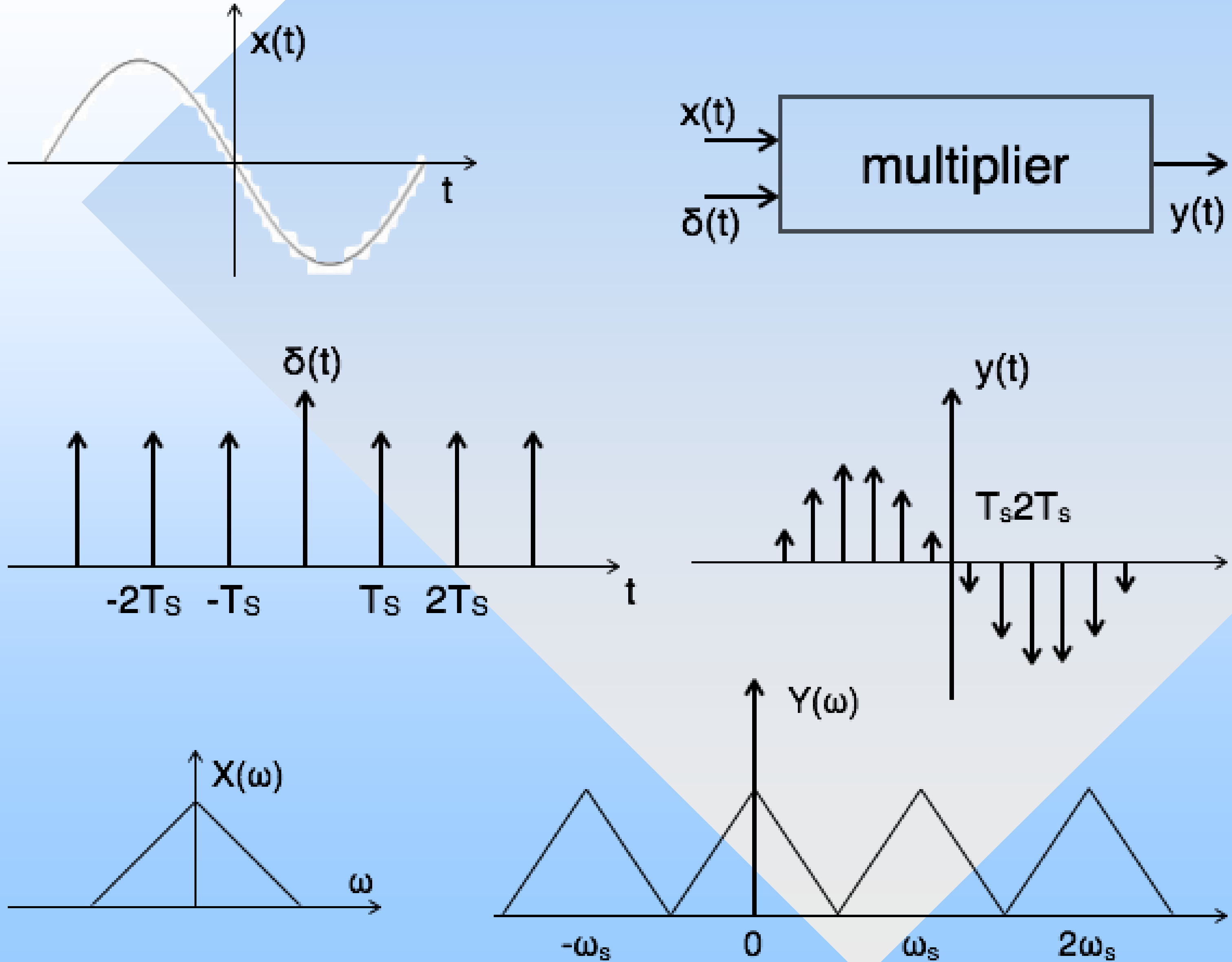
- Communication channel:
  - Has got a definite bandwidth, or
  - Has been given a limited bandwidth.
- While, spectrum of the sampled signal extends to infinity. This introduces crosstalk:
  - Due to the high frequency cutoff of channel.
  - Due to the low frequency cutoff of channel
- How to avoid? by instantaneous sampling

Definite Bandwidth





Sampling Output Spectrum



High Frequency Cutoff

- **Model**: Channel as an RC circuit LPF.
- Its upper frequency 3dB cutoff:

$$f_{High\ Frequency\ Cutoff} = f_{LPF} = \frac{1}{2\pi RC}$$

- Significant: only between **adjacent time slots**.
- Minimize: Channel time constant $\tau_c = RC$ is very small compared to guard interval:

$$\tau_c \ll \tau_g$$

Model
High Frequency Cutoff of the Channel

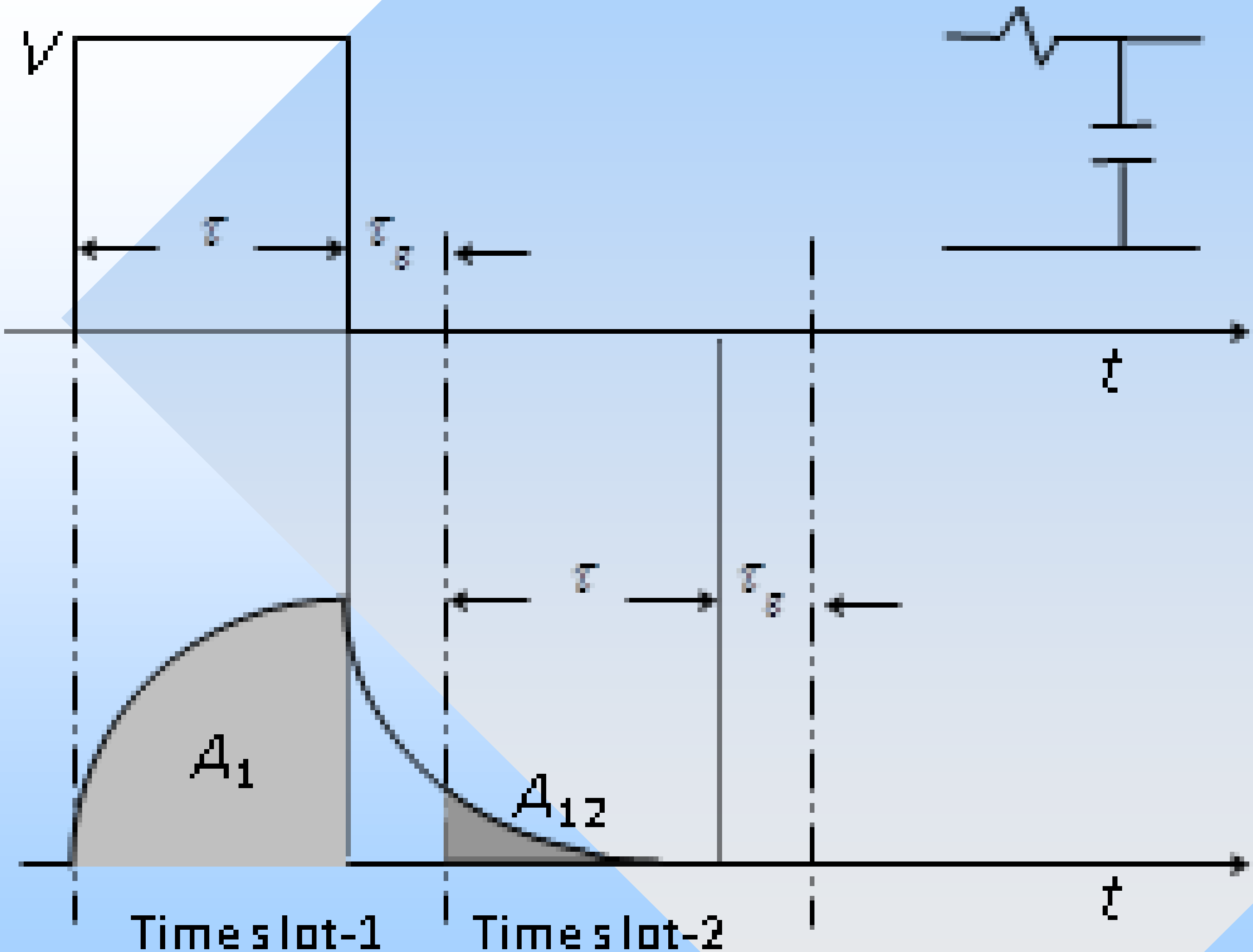


Fig.1.10. Crosstalk due to the Upper Cutoff

High Crosstalk Factor

- If channel 2 is empty and input pulse is of value *V*, then the crosstalk factor is given:

$$A_1 \cong \tau V$$

$$A_{12} \cong \tau_c V e^{-\frac{\tau_g}{\tau_c}}$$

$$K_{High\ Frequency\ Cotoff} = \frac{A_{12}}{A_1} = \frac{\tau_c}{\tau} e^{-\frac{\tau_g}{\tau_c}}$$

Low Frequency Cutoff

- **Model**: Channel as an RC HPF circuit.
- Its low frequency 3dB cutoff:

$$f_{Low\ Frequency\ Cutoff} = f_{HPF} = \frac{1}{2\pi RC}$$

- Sample pulse in TS_1 develops exponential *tilt* in TS_2 given by:

$$tilt = \Delta = V\left(1 - e^{-\frac{\tau}{\tau_c}}\right)$$

- **Significant**: Extends to several time slots.

Model
Low Frequency Cutoff of the Channel

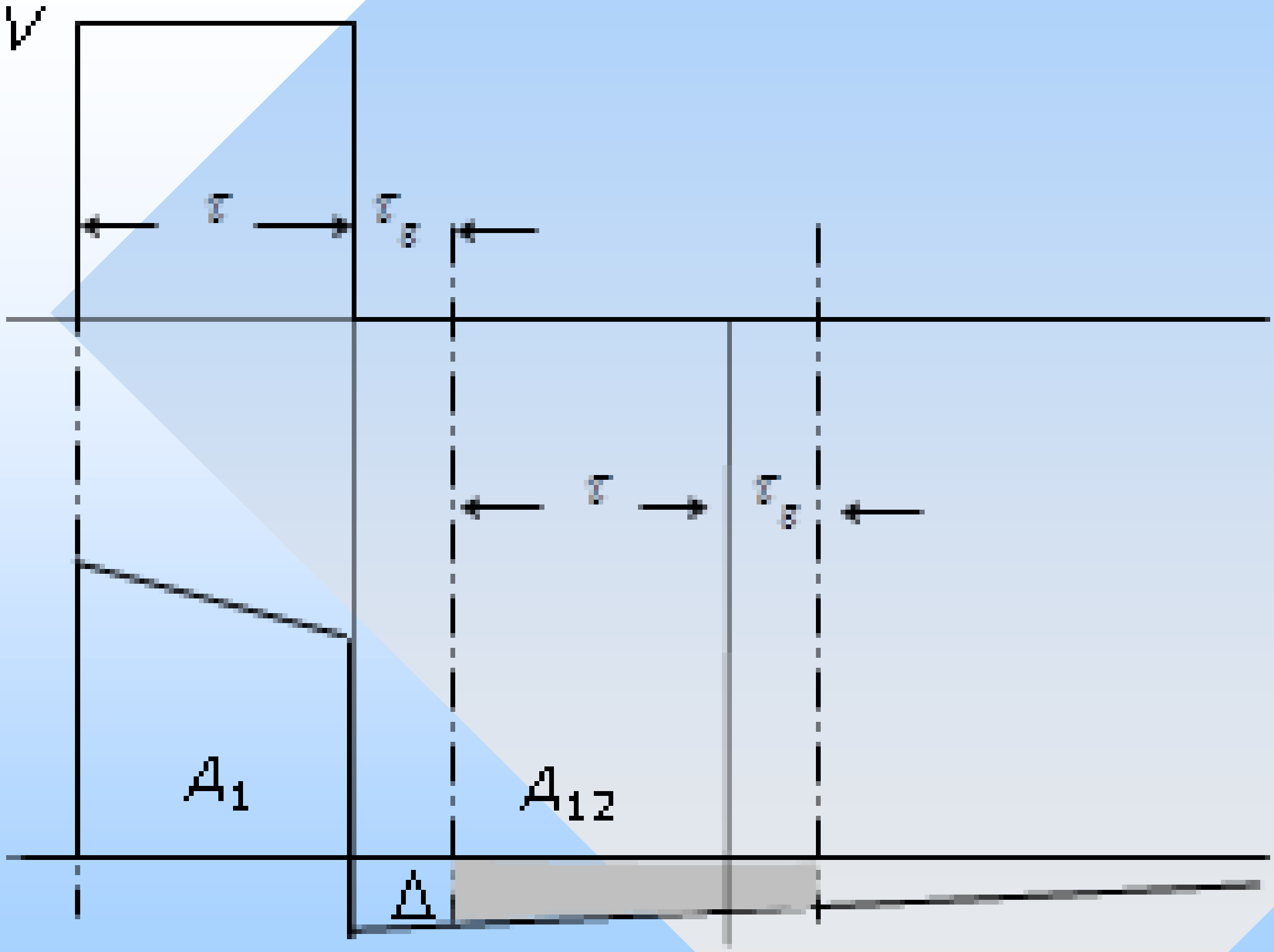


Fig.1.11. Crosstalk due to the Lower Cutoff

Low Crosstalk Factor

- Minimize: When $\tau \ll \tau_c$ the *telt* Δ could be linearly approximated to:

$$\Delta = \frac{\tau}{\tau_c} V$$

- When $\tau_g \ll \tau_c$, then:

$$A_1 \cong \tau V$$

$$A_{12} = \tau\Delta = \frac{\tau^2 V}{\tau_c}$$

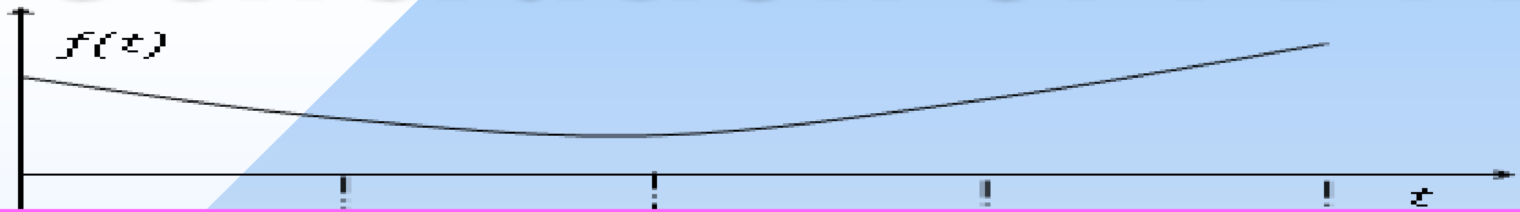
$$K_{Low\ Frequency\ Cutoff} = \frac{\tau}{\tau_c}$$

Pulse Time Modulation

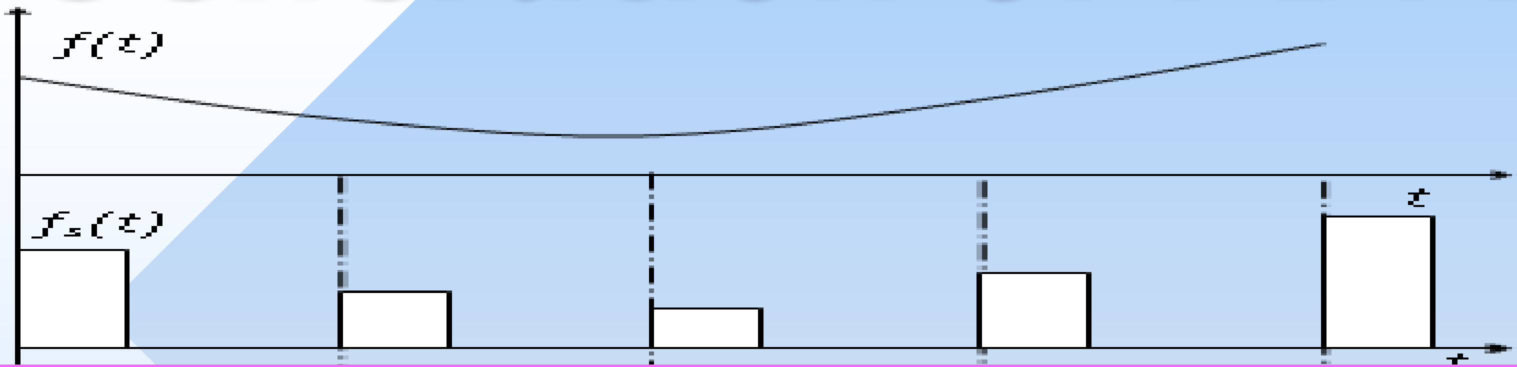
Pulse Time Modulation

- Pulse Width Modulation
  - PDM or PWM.
- Pulse Position Modulation
  - PPM

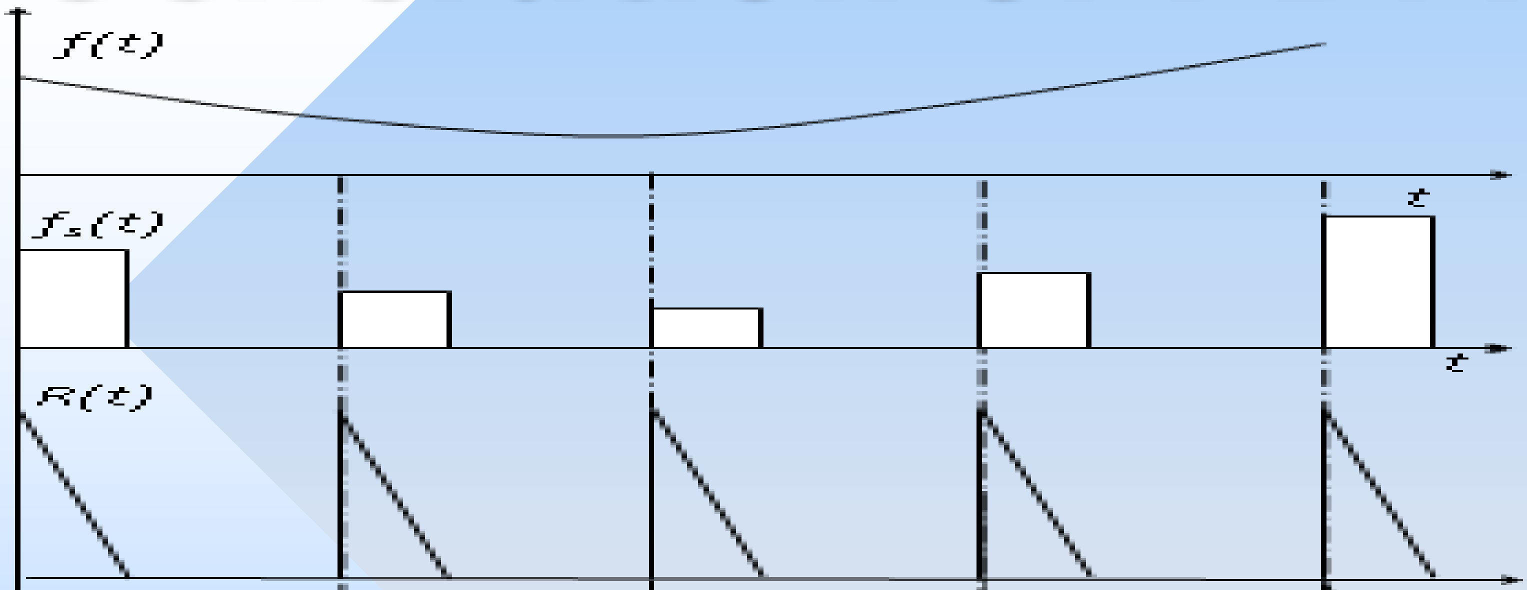
Generation of PDM



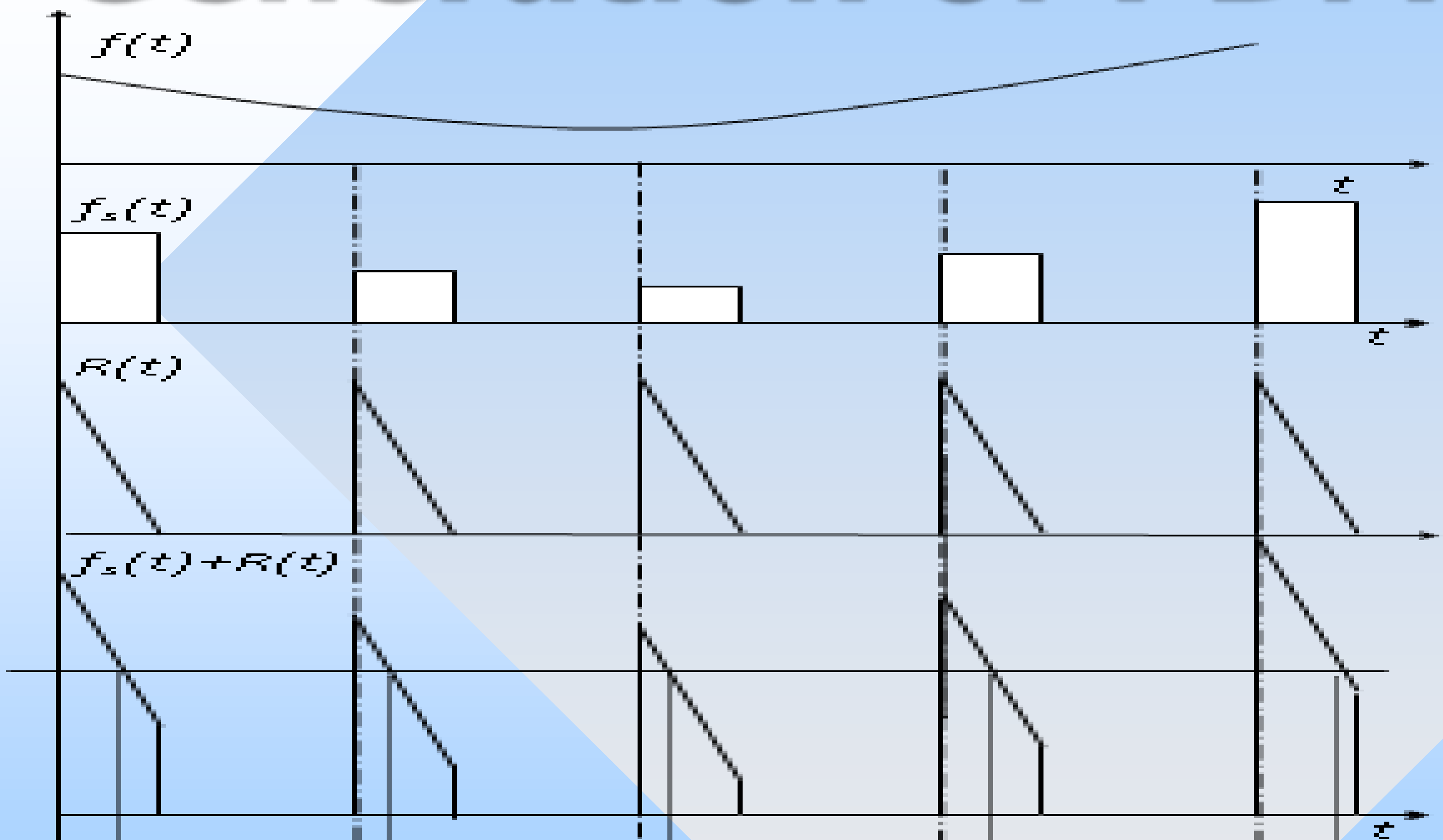
Generation of PDM



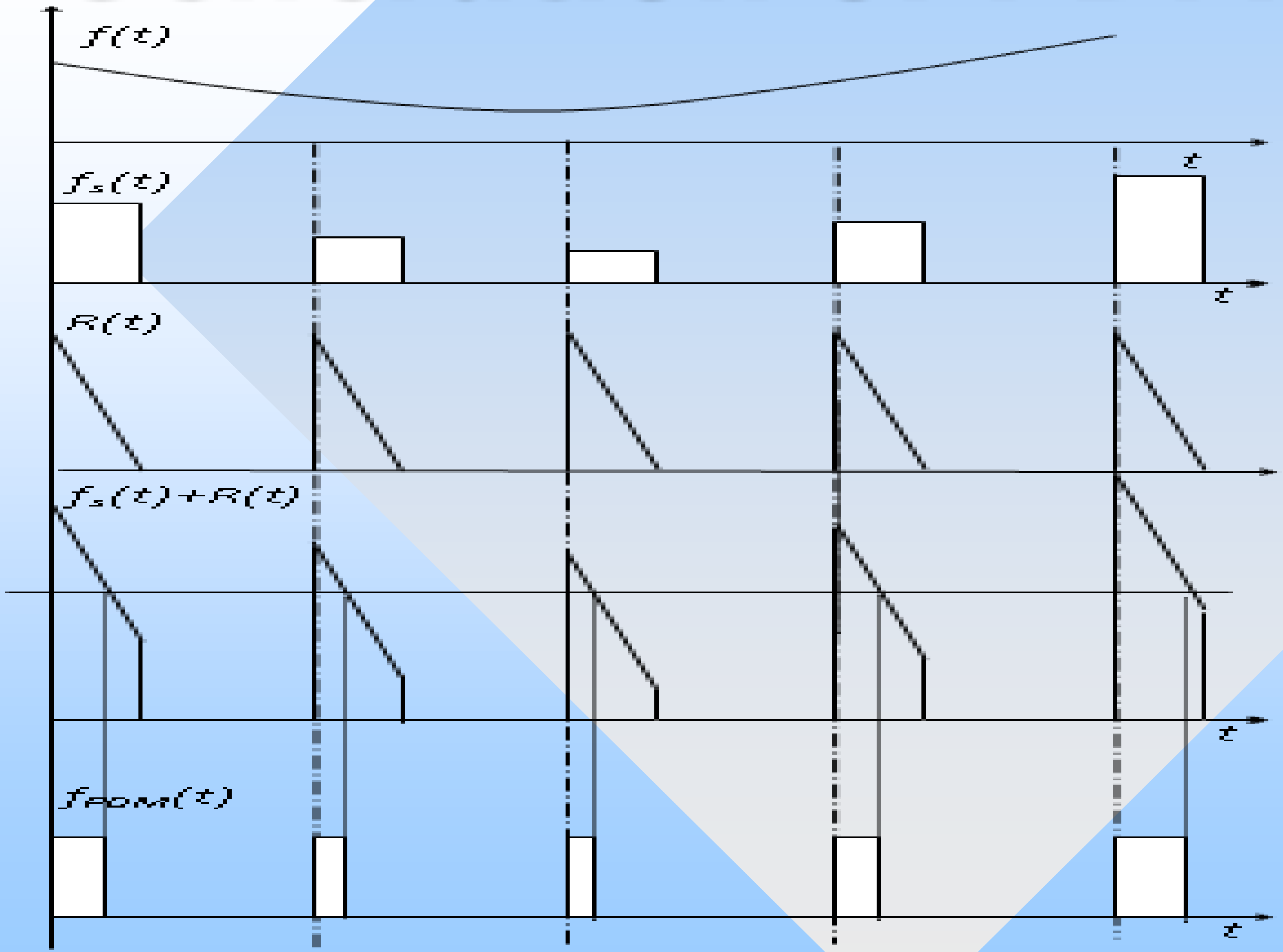
Generation of PDM



Generation of PDM



Generation of PDM

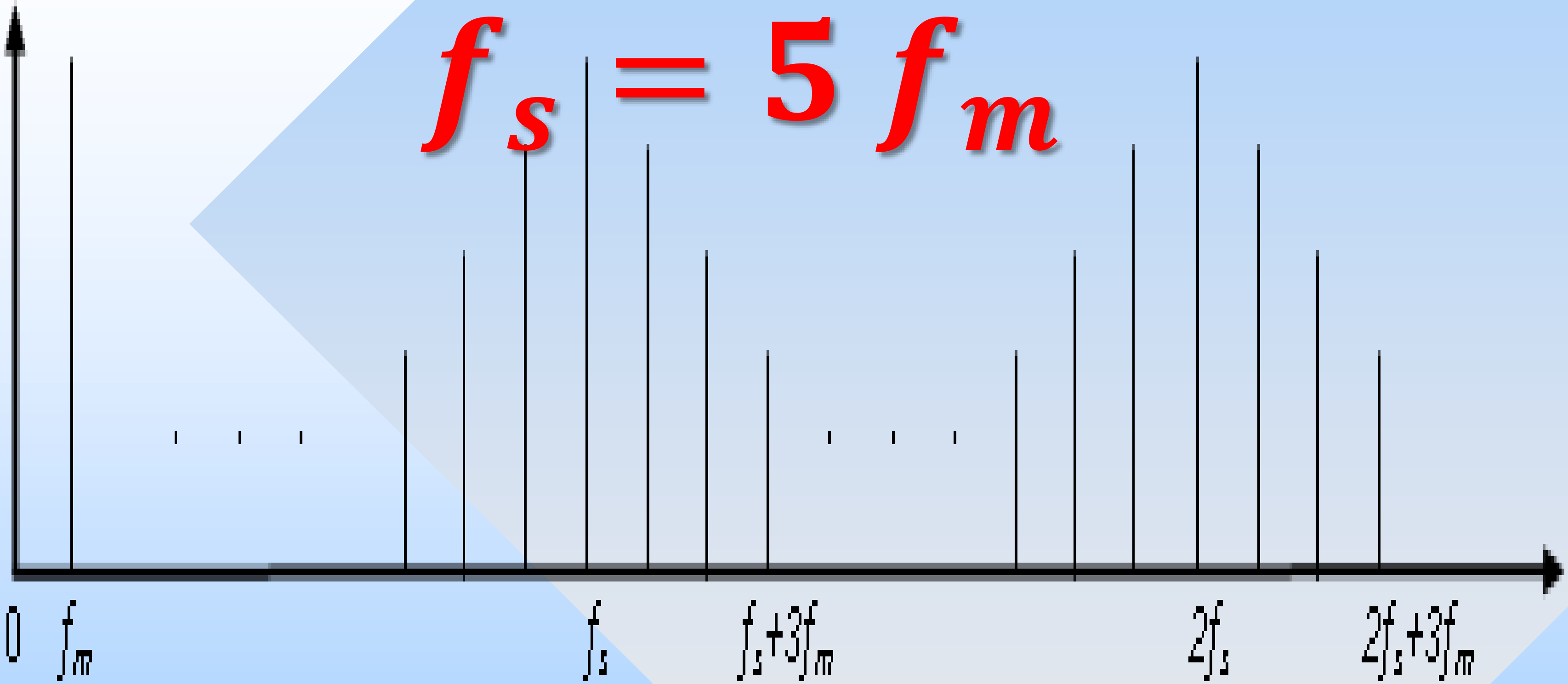


Recovery PDM, PPM

- Using LPF Only
  - When $f_s \gg 2f_m$
- First:
  - First: Conversion to PAM
  - Second: Using LPF

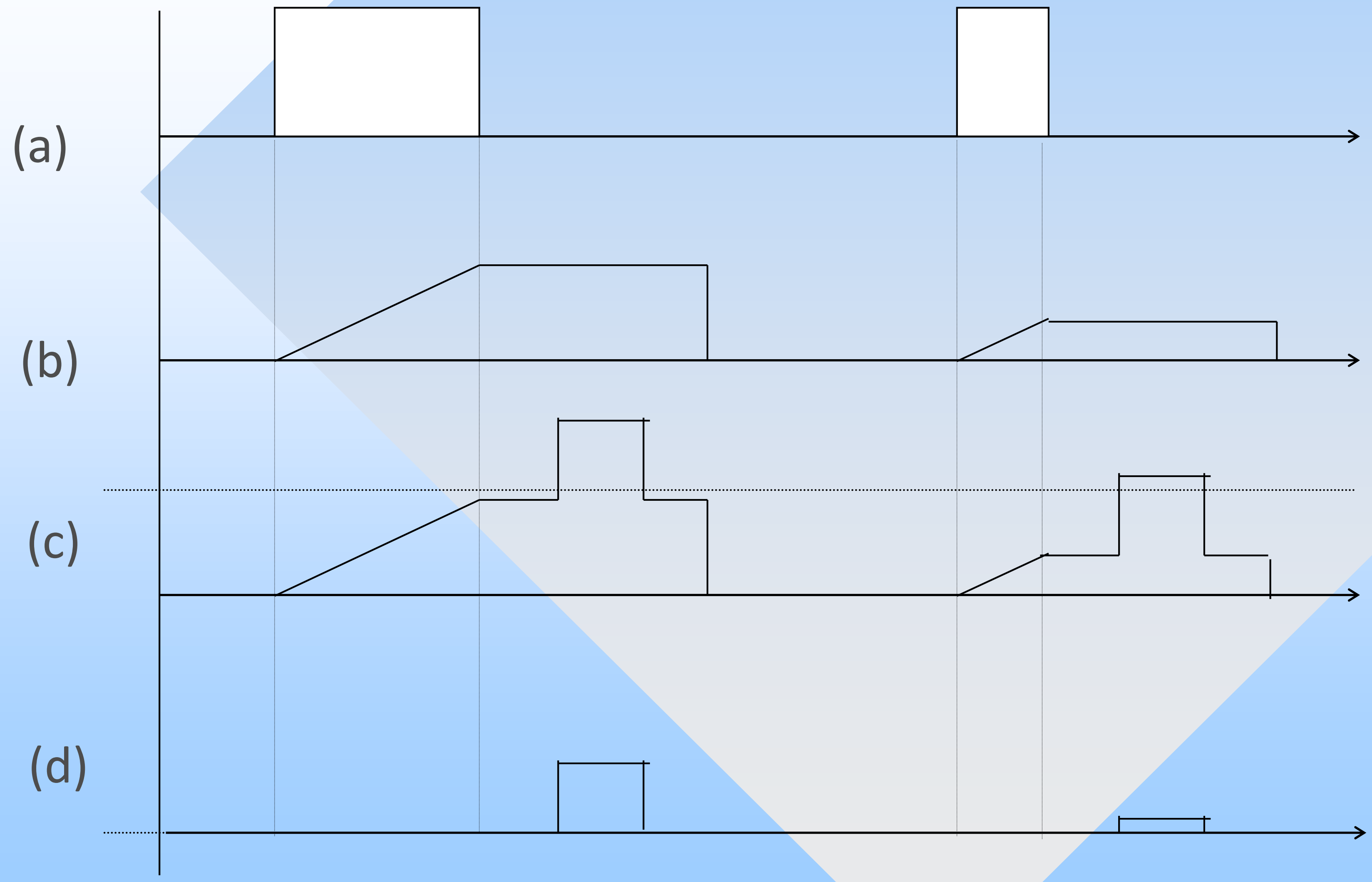
Spectrum of PDM

$$f_s = 5\,f_m$$



PDM Sampled with sinusoidal $f(t)$ where modulation superimposed on the trailing edge of the pulse.

Conversion to PAM



Digital Pulse Modulation

Advantages of Digital Modulation

- Information is only conveyed be the presence or absence of a pulse.
  - Pulse shape or exact amplitude is not significant.
- Signals be correct regenerated on transmission.
- Ability of using all digital circuitry.
- Applicable to all forms of signal processing.
- Noise and interference can be minimized using coding techniques.

Digital Pulse Modulation

- Delta Modulation
  - Delta.
- Adaptive Delta Modulation
  - Adaptive Delta

Delta Modulation

- DM is a digital technique in which message is encoded into sequence of binary symbols.
- It is a simple 1-bit PCM code to achieve digital transmission.
- Operation depends on whether the sample is larger or smaller than previous one.
- Circuits to implement both modulator and demodulator are extremely simple.

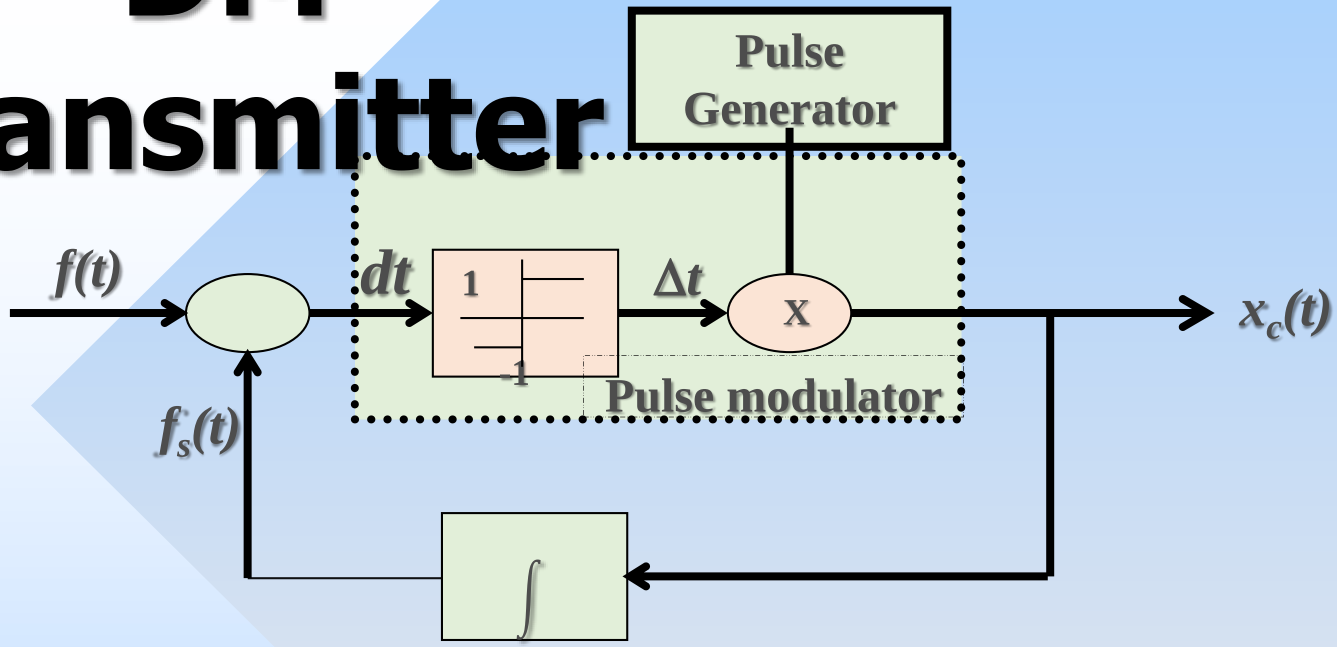
DM Transmitter

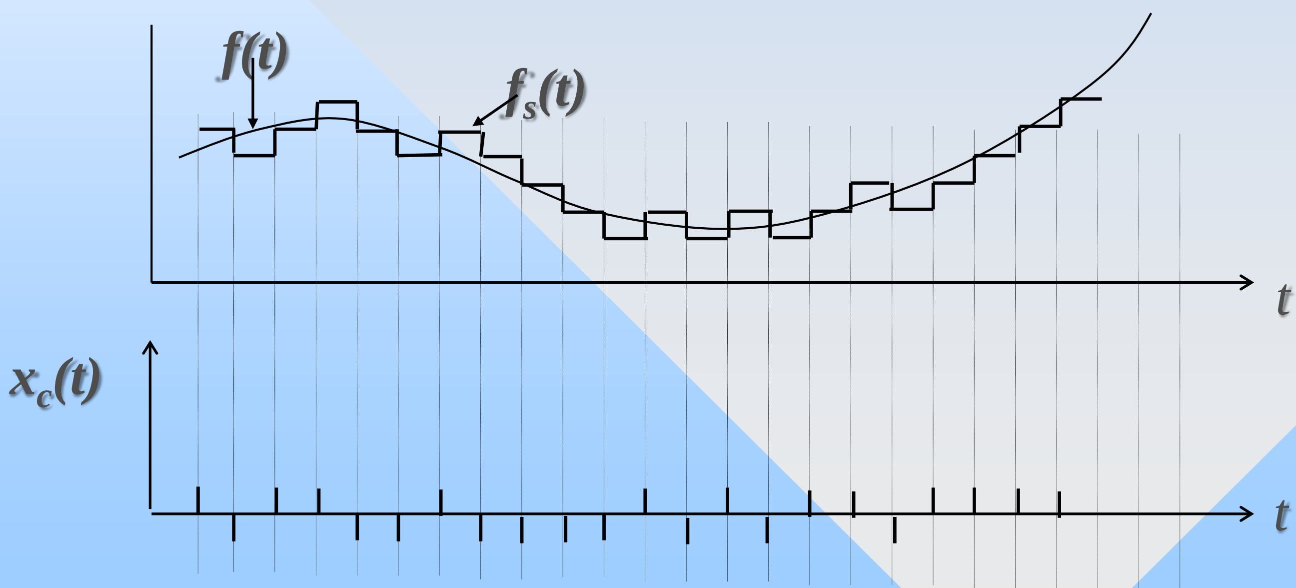
- $f_s(t)$ is a stair step approximation of $f(t)$.
- Modulator input is: $d(t) = f(t) - f_s(t)$
- $d(t)$ is hard-limited and multiplied by the pulse generator to yield the output:

$$x_c(t) = \Delta(t)\sum_{-\infty}^{\infty}\delta(t - nT_s) = \sum_{-\infty}^{\infty}\Delta(nT_s)\delta(t - nT_s)$$

$$f_s(t) = \int_0^t x_c(t)\,dt = \sum_{n-\infty}^{\infty}\Delta(nT_s)\int_0^t \delta(y - nT_s)\,dy$$

DM Transmitter



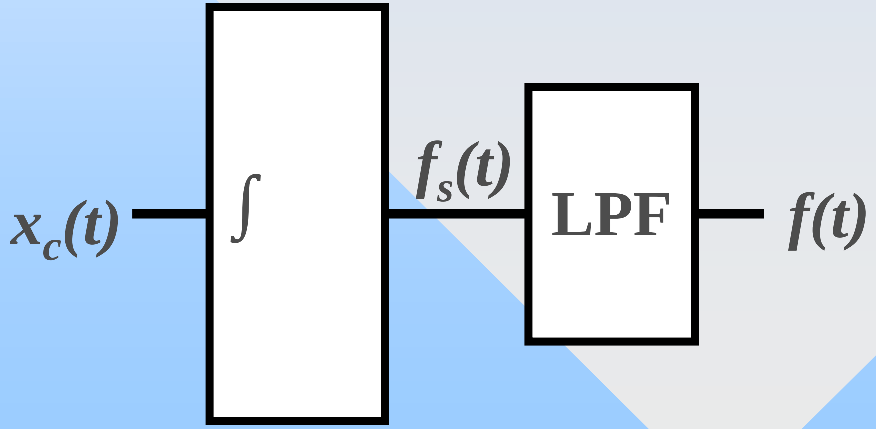


DM Receiver

Demodulation of DM is simple by:

- Integrating $x_c(t)$ to form stair approximation $f_s(t)$
- That is low pass filtered to suppress discrete jumps.

Since LPF approximates an integrator, it is often possible to eliminate the integrator.

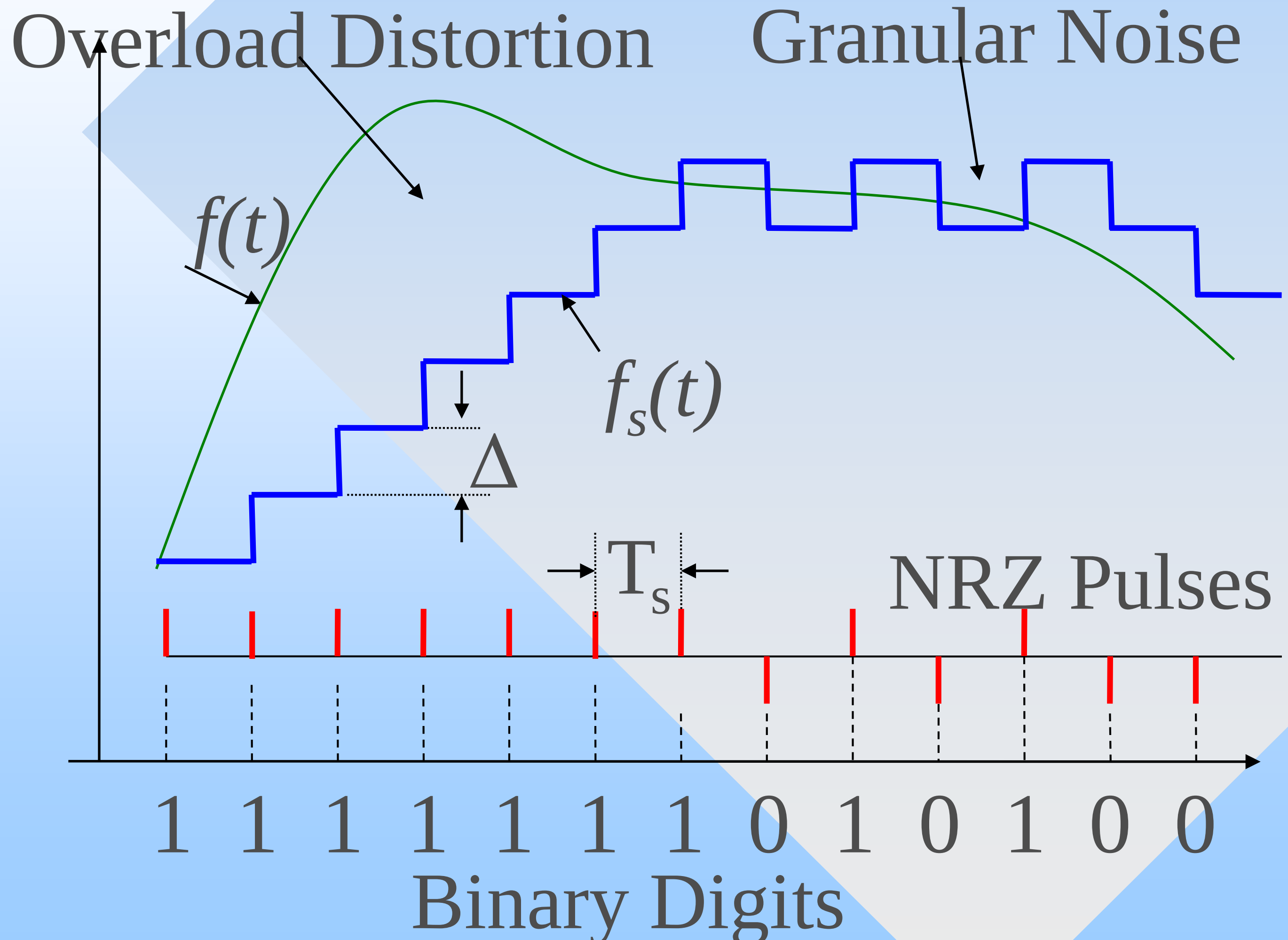


Granular Noise

When changes in $f(t)$ are less than the step size, DM no longer follows the signal and it produces a train of alternating positive and negative pulses:

- Similar to quantization noise in conventional PCM.
- When $f(t)$ has constant amplitude reconstructed signal has variations that were not present in the original signal.
- It can be reduced by decreasing the step size (**high resolution**).

Drawbacks of DM



Slope Overload Distortion

When the analogue signal changes at a faster rate [i.e, $f(t)$ slope is too high].

Its slope is greater than DM can maintain.

Can be reduced by:

- ❑ Increasing the clock frequency.
- ❑ Increasing the step size (**Low resolution**)

Adaptive DM Transmitter

If $f(t)$ is relatively constant, the pulses $x_c(t)$ will alternate sign.

So the dc value of LPF output is nearly zero

(**minimum step size**)

If $f(t)$ increases or decreases rapidly, $x_c(t)$ will have same polarity over that period.

So the dc value of LPF output is large

(**larger step size**)

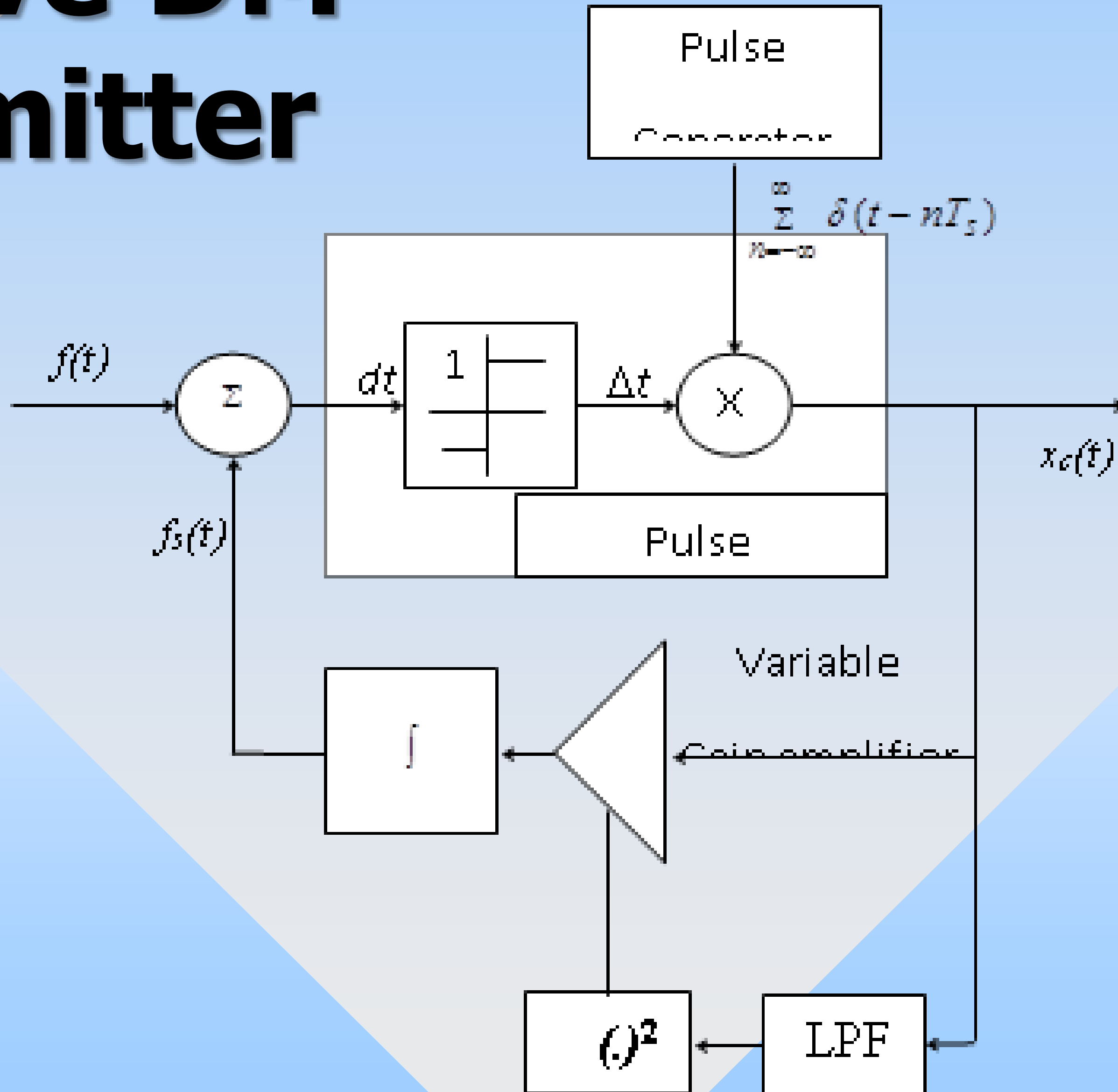


Fig.1.22 Adaptive Delta Modulation

Adaptive DM Receiver

The receiver of ADM should be adaptive also